ESE 531: Digital Signal Processing

Lec 23: April 19, 2018 Wavelet Transform, Compressive Sensing



Penn ESE 531 Spring 2018 – Khanna



Today

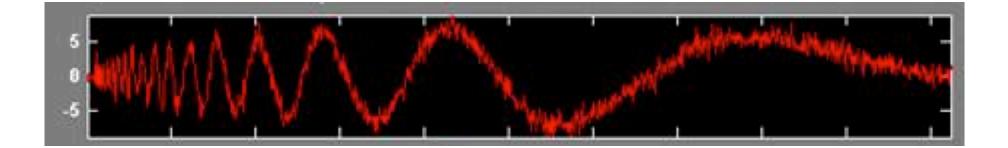
- Wavelet Transform
- Compressive Sampling/Sensing

Wavelet Transform



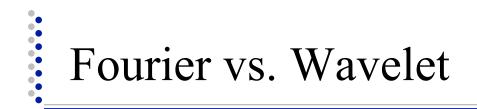


 Some signals obviously have spectral characteristics that vary with time

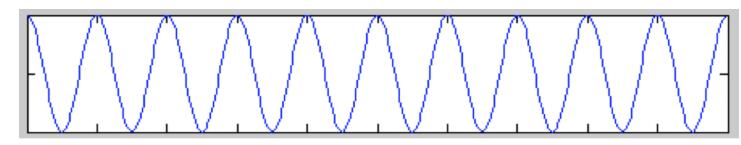


Criticism of Fourier Spectrum

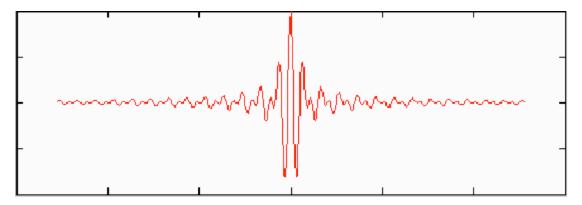
- It's giving you the spectrum of the 'whole timeseries'
- Which is OK if the time-series is stationary. But what if its not?
- We need a technique that can "march along" a time series and that is capable of:
 - Analyzing spectral content in different places
 - Detecting sharp changes in spectral character

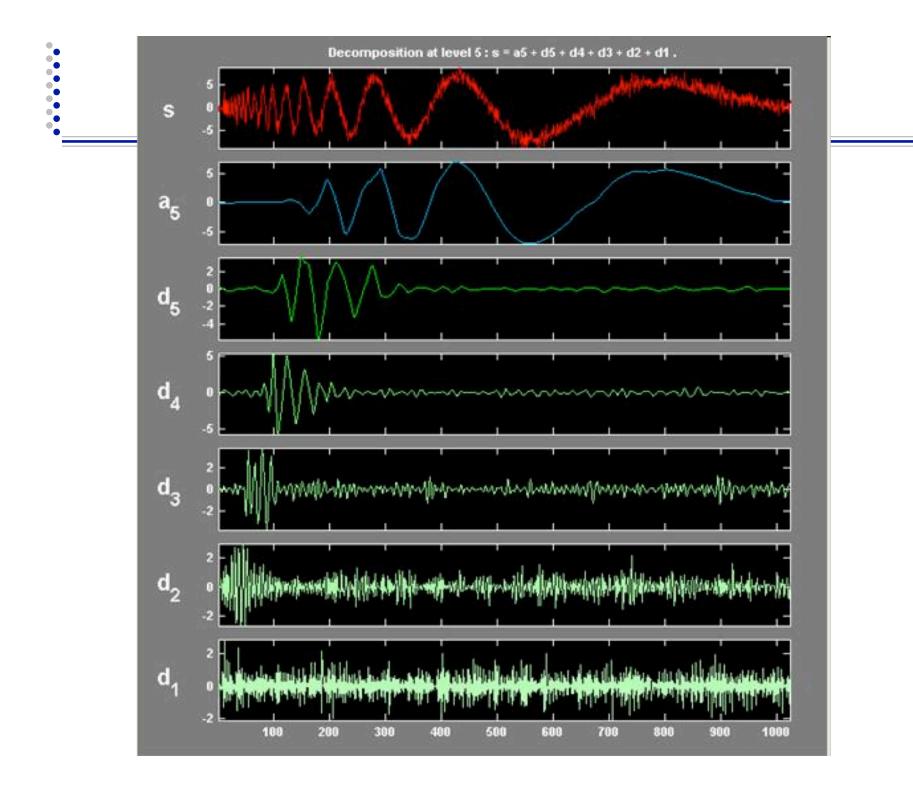


 Fourier Analysis is based on an indefinitely long cosine wave of a specific frequency



 Wavelet Analysis is based on an short duration wavelet of a specific center frequency







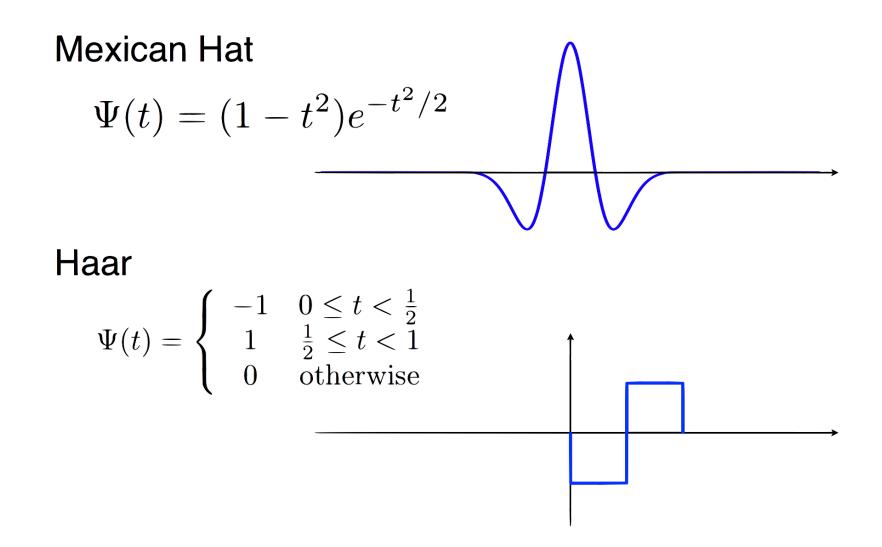
□ All wavelet derived from *mother* wavelet

$$\Psi_{s,\tau}(t) = \frac{1}{\sqrt{s}} \Psi\left(\frac{t-\tau}{s}\right)$$

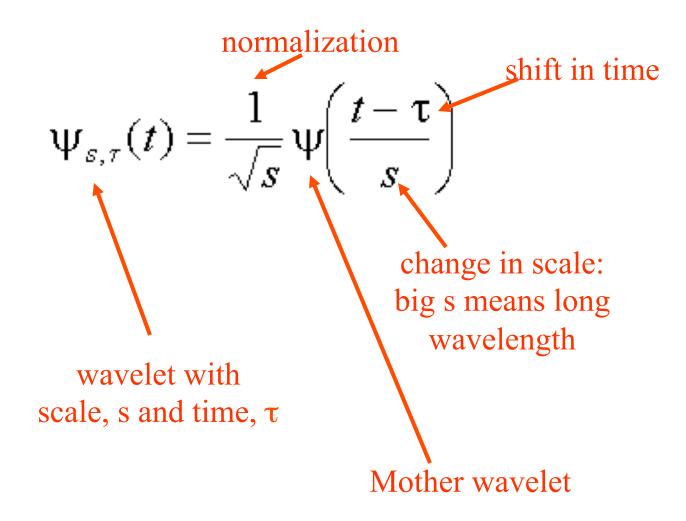
Must satisfy

$$\int_{-\infty}^{\infty} |\Psi(t)|^2 dt = 1 \quad \Rightarrow \text{ unit norm}$$



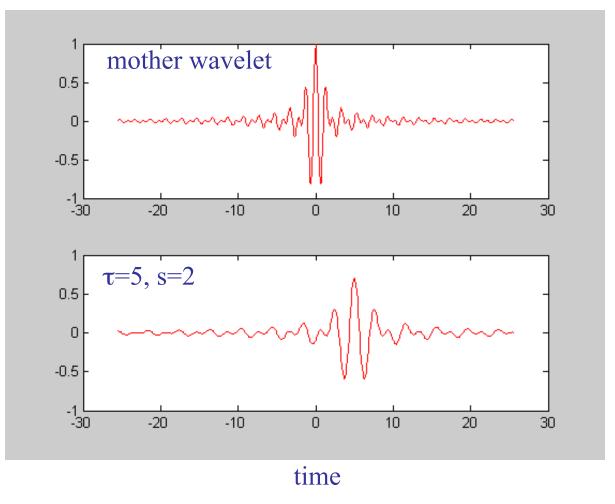






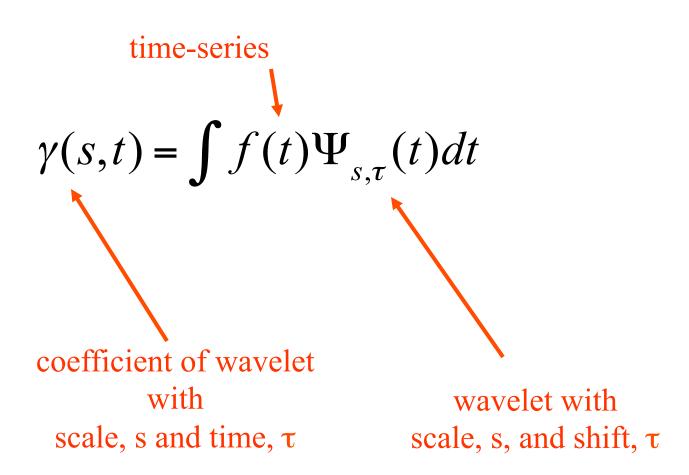


 $\Box \Psi(t) = 2 \operatorname{sinc}(2t) - \operatorname{sinc}(t)$



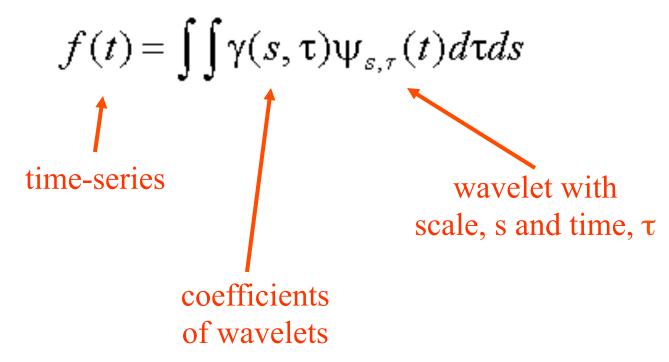
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Inverse Wavelet Transform

 Build up a time-series as sum of wavelets of different scales, s, and positions, t



Wavelet Transform

Determining the wavelet coefficients for a fixed scale, s, can be thought of as a filtering operation

$$\gamma(s,t) = \int f(t) \Psi_{s,\tau}(t) dt$$

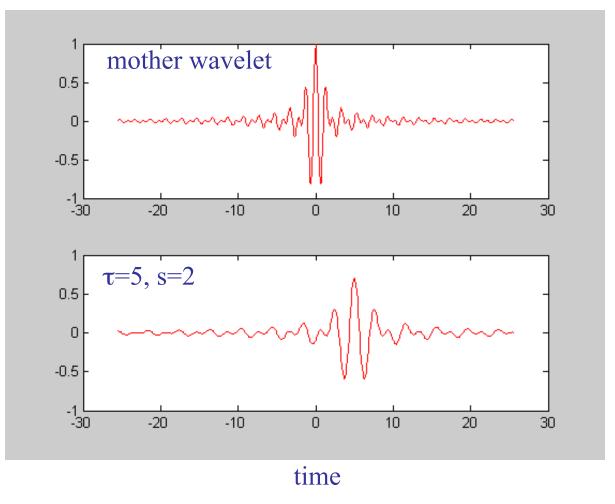
$$\gamma_s(t) = \int f(t) \Psi_s(t) dt = f(t) * \Psi_s(t)$$

• where

$$\Psi_{s}(t) = \frac{1}{\sqrt{s}} \Psi(\frac{t}{s})$$

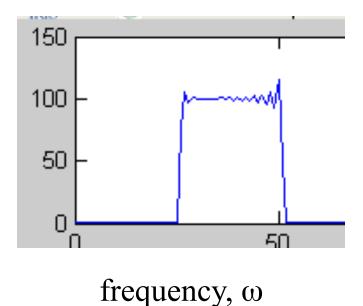


 $\Box \Psi(t) = 2 \operatorname{sinc}(2t) - \operatorname{sinc}(t)$



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Wavelet coefficients are a result of bandpass

filtering

Discrete wavelets:

□ Scale wavelets only by powers of 2

- $s_j = 2^j$
- And shifting by multiples of s_j for each successive scale

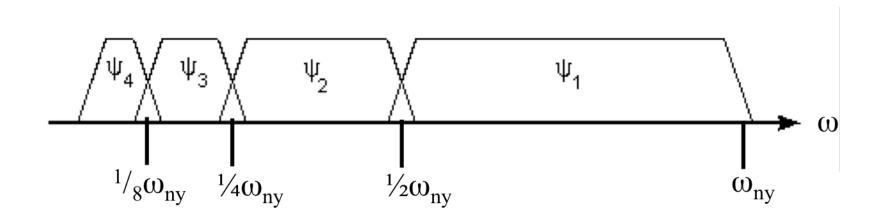
•
$$\tau_{j,k} = 2^{j}k$$

- $\Box \text{ Then } \mathbf{Y}(\mathbf{S}_{j}, \mathbf{T}_{j,k}) = \mathbf{Y}_{jk}$
 - where $j = 1, 2, ..., \infty, k = -\infty ... -2, -1, 0, 1, 2, ..., \infty$

$$\gamma_{j,k} = \frac{1}{\sqrt{2^j}} \int f(t) \Psi\left(\frac{t - k2^j}{2^j}\right) dt$$

Discrete Wavelet Transform

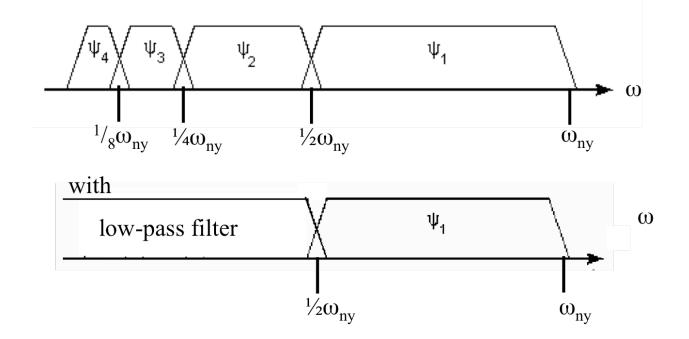
 The factor of two scaling means that the spectra of the wavelets divide up the frequency scale into octaves (frequency doubling intervals)



Discrete Wavelet Transform

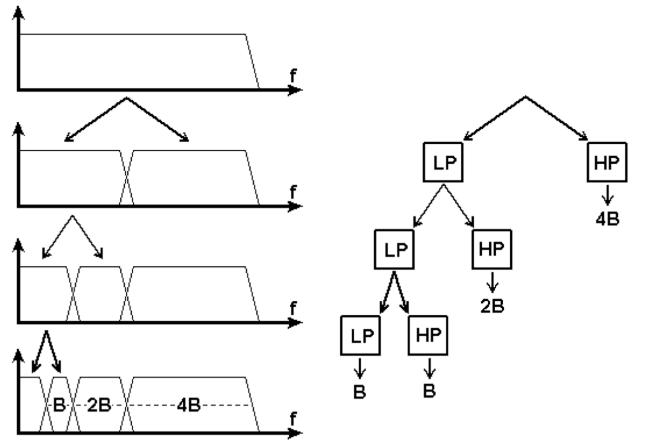
As we saw previously, the coefficients of Ψ is just the band-pass filtered time-series, where Ψ is the wavelet, now viewed as the impulse response of a bandpass filter. Discrete Wavelet Transform

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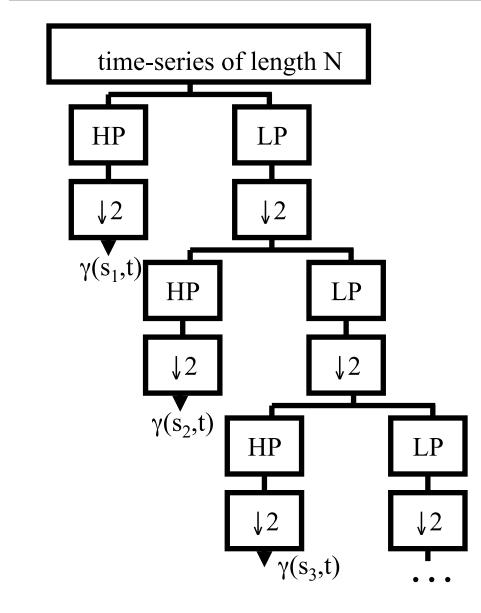




□ Repeat recursively!





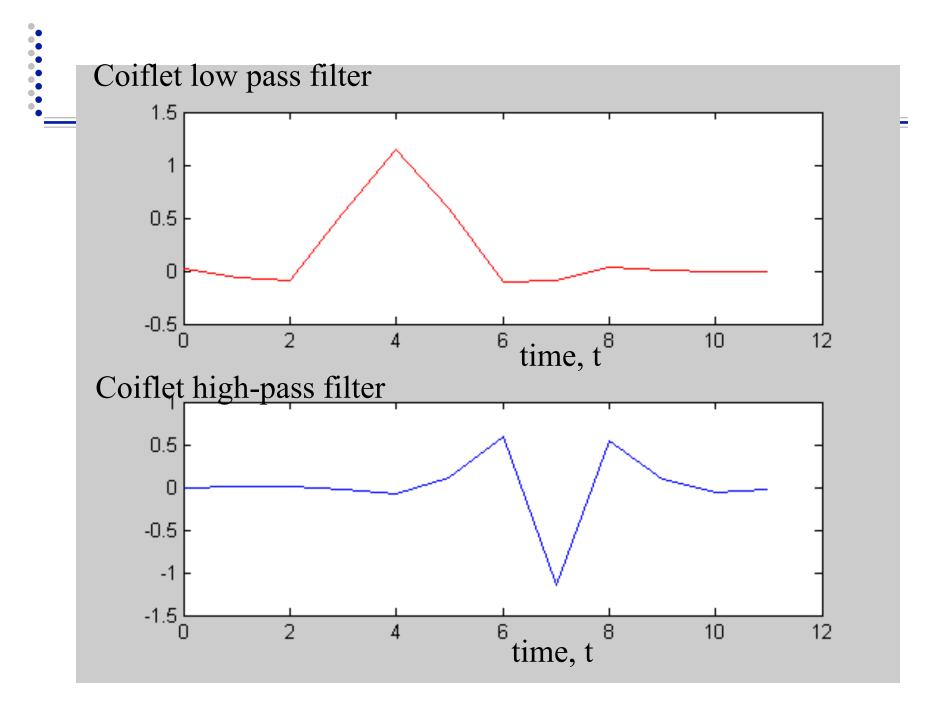


 $\gamma(s_1,t)$: N/2 coefficients

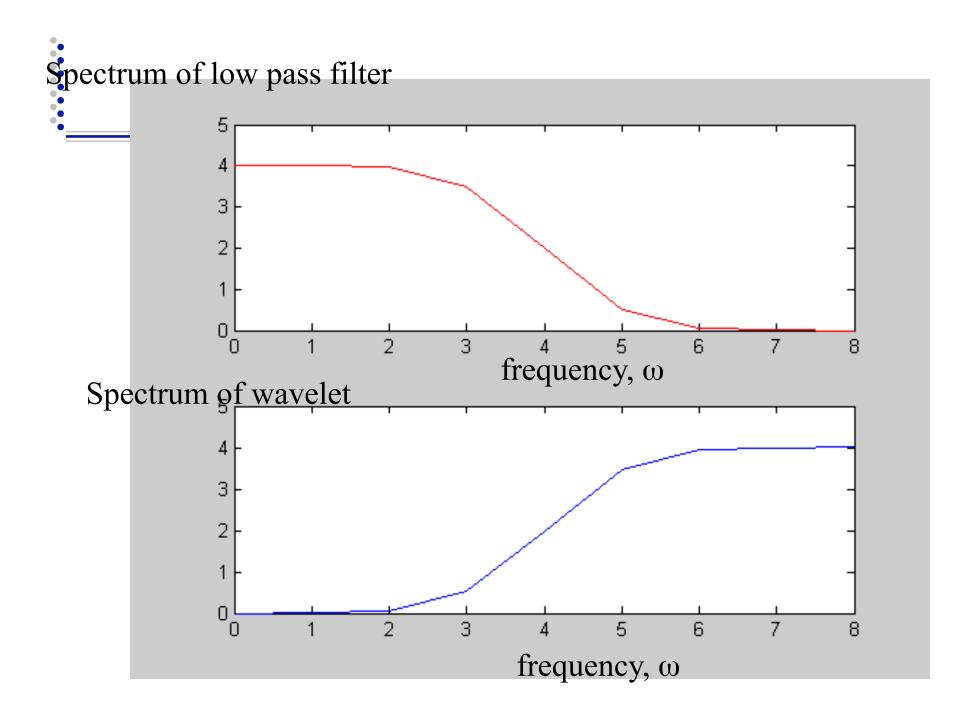
 $\gamma(s_2,t)$: N/4 coefficients

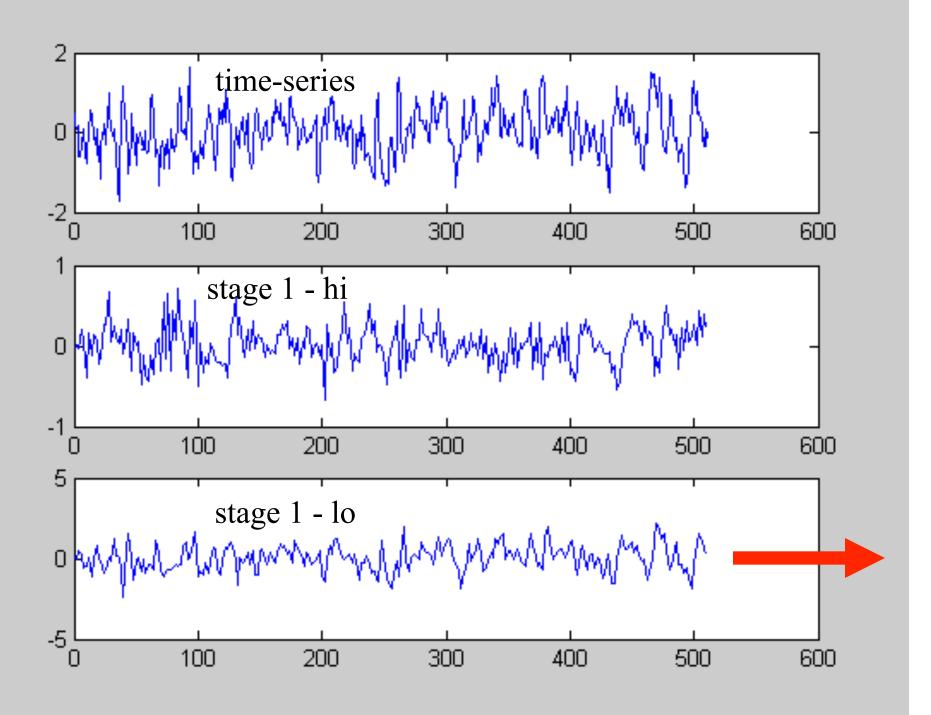
 $\gamma(s_2,t)$: N/8 coefficients

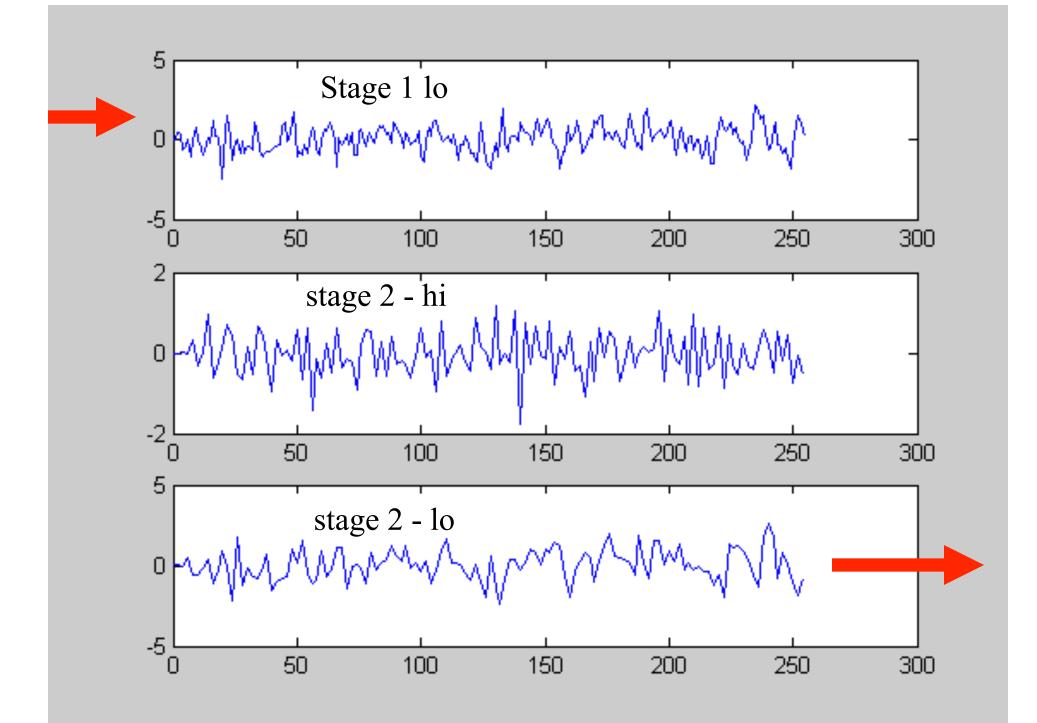
Total: N coefficients

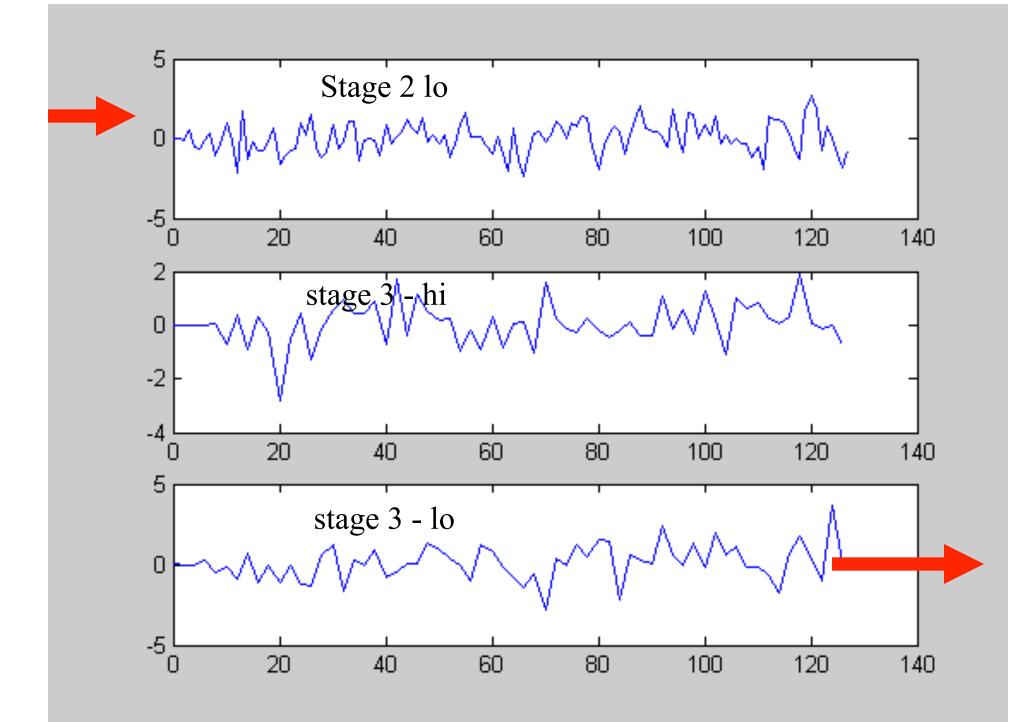


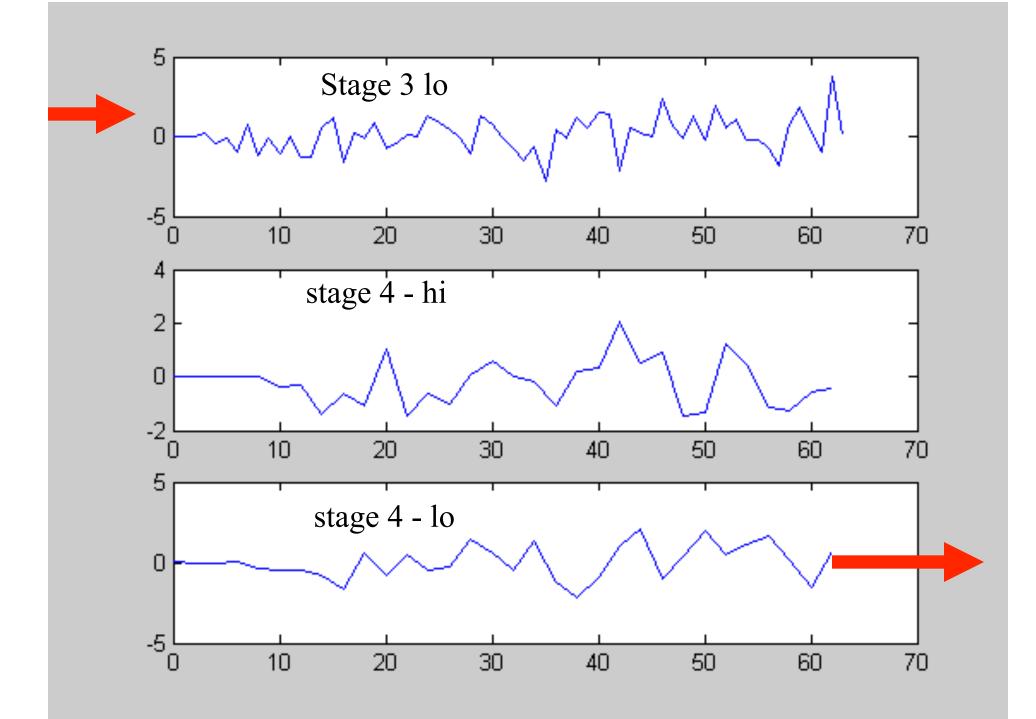
From http://en.wikipedia.org/wiki/Coiflet

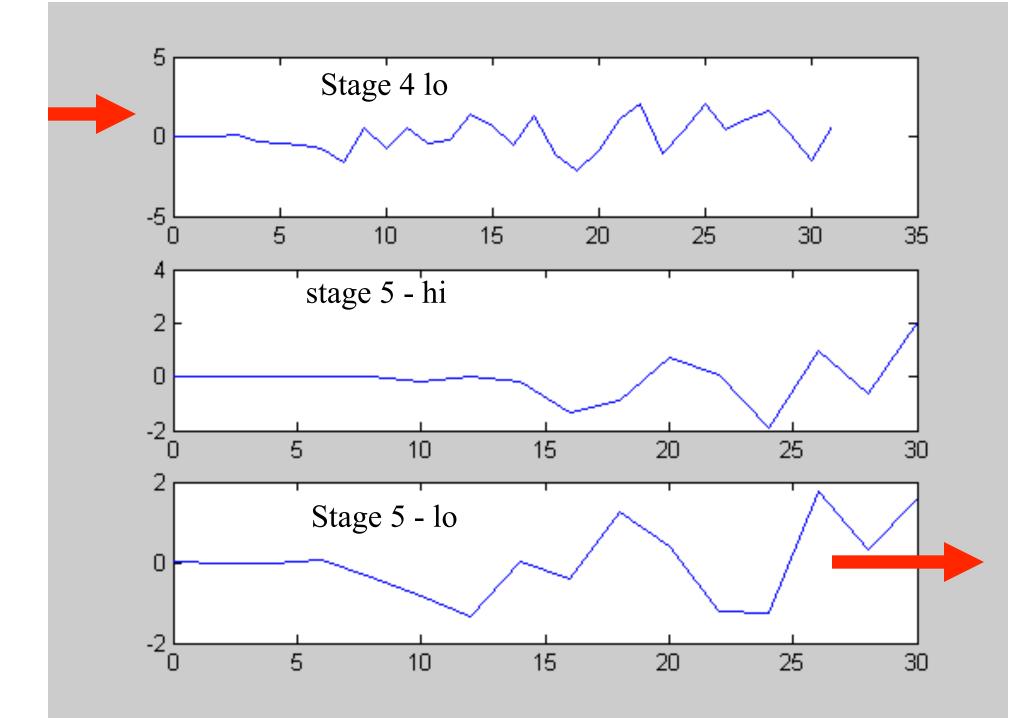


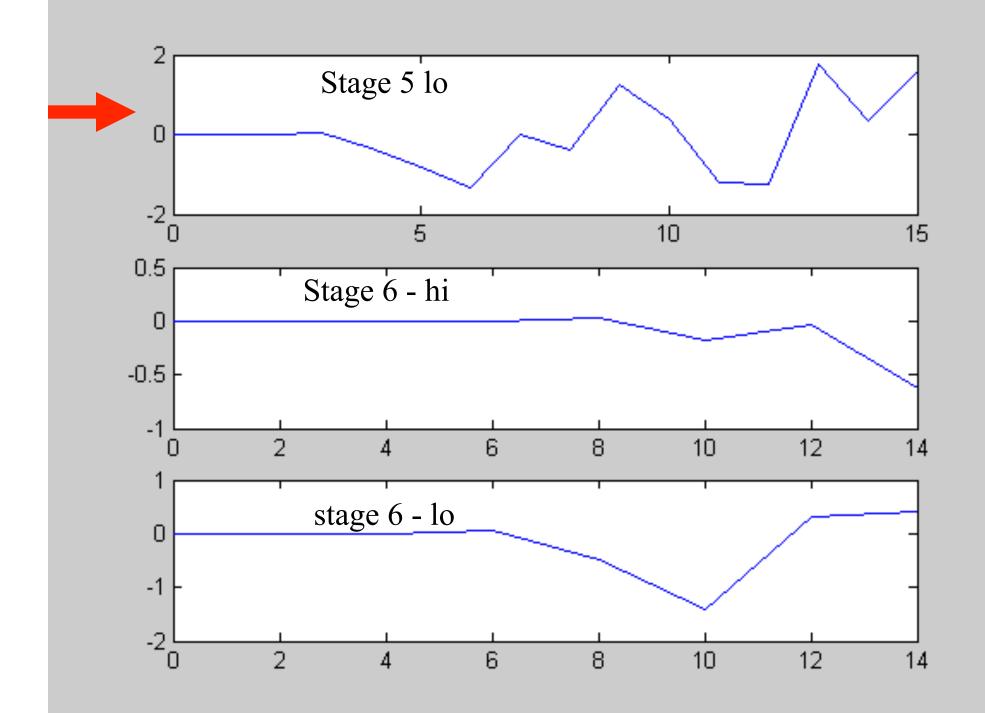


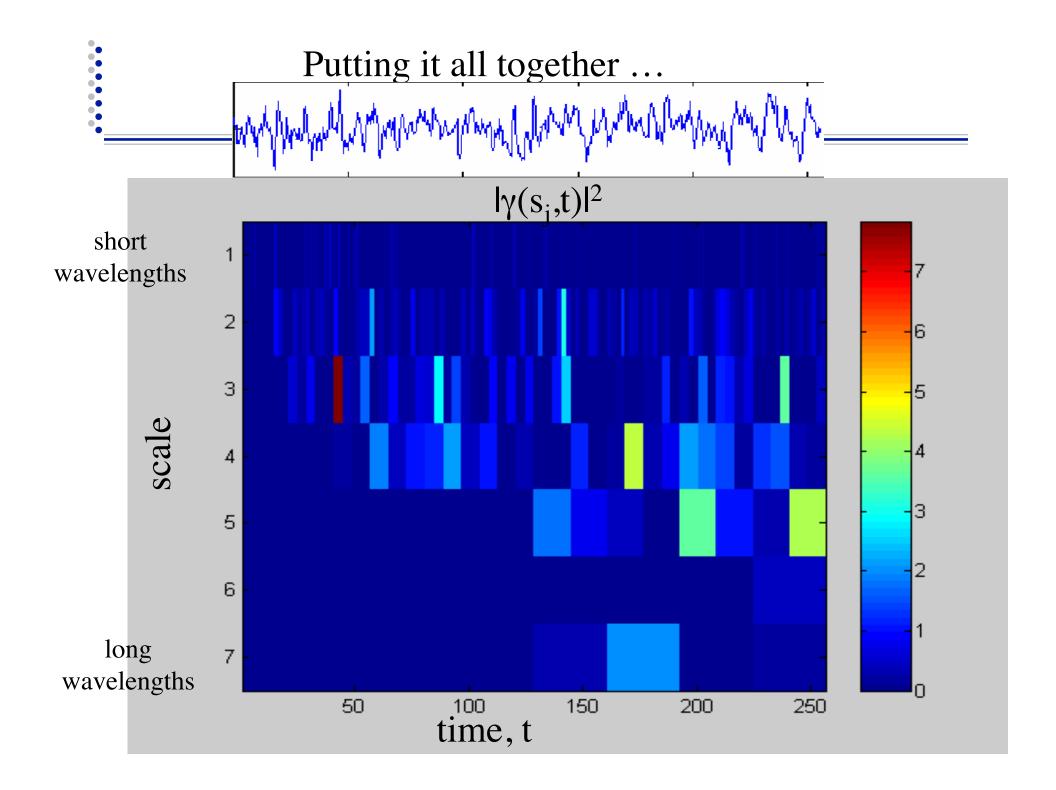






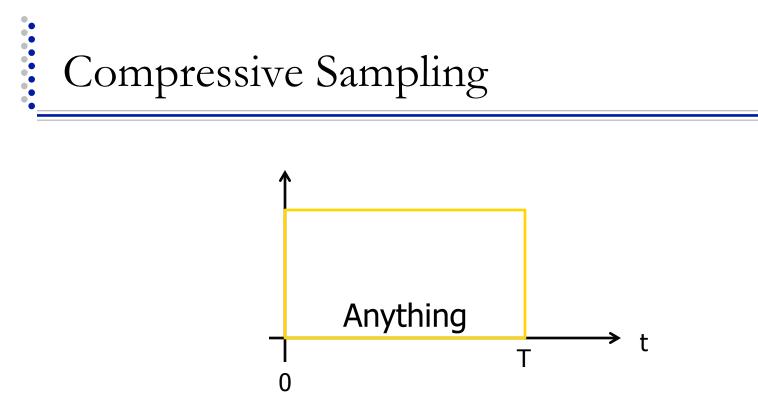




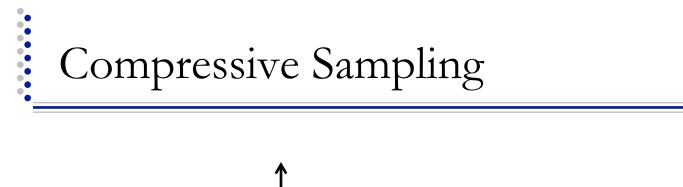


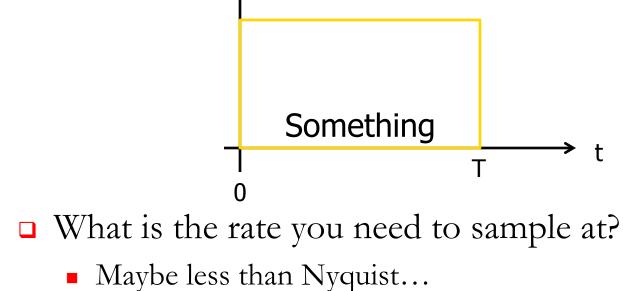
Compressive Sampling





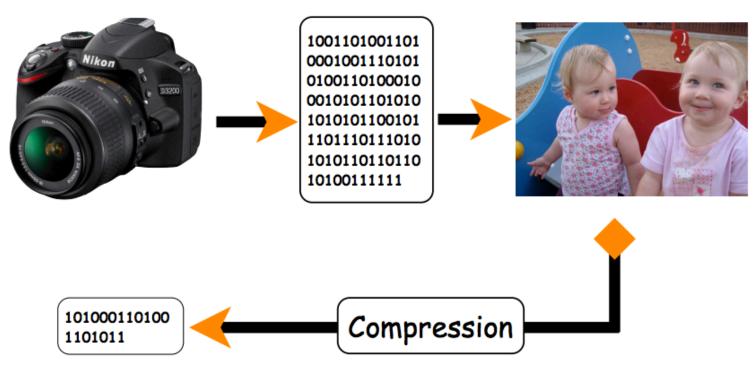
- □ What is the rate you need to sample at?
 - At least Nyquist







- □ Standard approach
 - First collect, then compress
 - Throw away unnecessary data





Examples

- Audio 10x
 - Raw audio: 44.1kHz, 16bit, stereo = 1378 Kbit/sec
 - MP3: 44.1kHz, 16 bit, stereo = 128 Kbit/sec
- Images 22x
 - Raw image (RGB): 24bit/pixel
 - JPEG: 1280x960, normal = 1.09bit/pixel
- Videos 75x
 - Raw Video: (480x360)p/frame x 24b/p x 24frames/s + 44.1kHz x 16b x 2 = 98,578 Kbit/s
 - MPEG4: 1300 Kbit/s

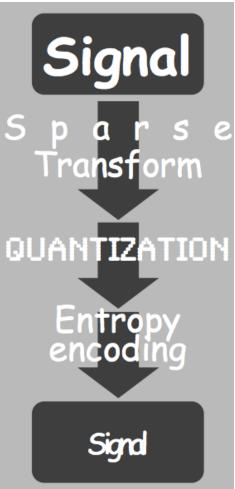


- Almost all compression algorithm use transform coding
 - mp3: DCT
 - JPEG: DCT
 - JPEG2000: Wavelet
 - MPEG: DCT & time-difference

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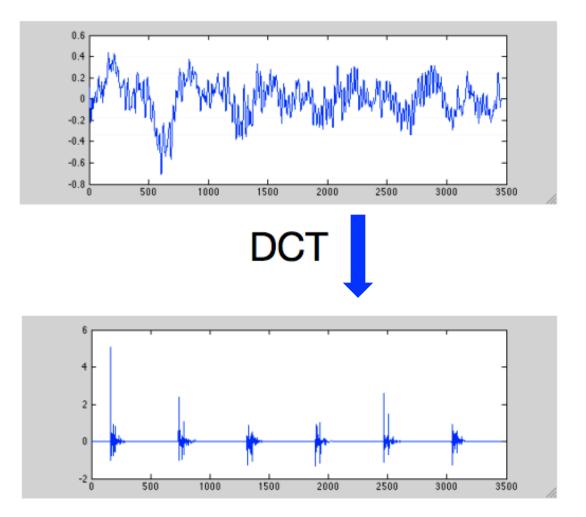


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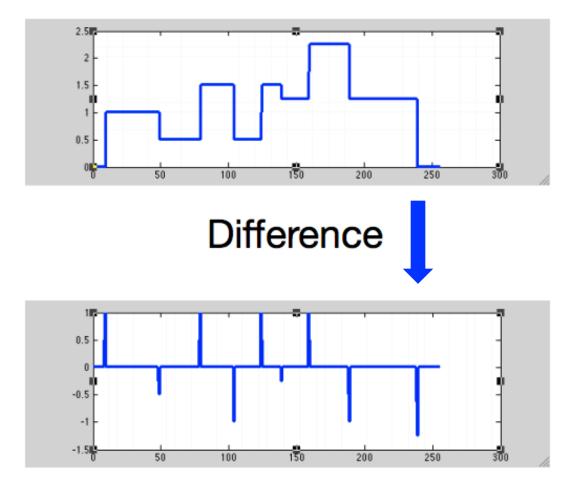


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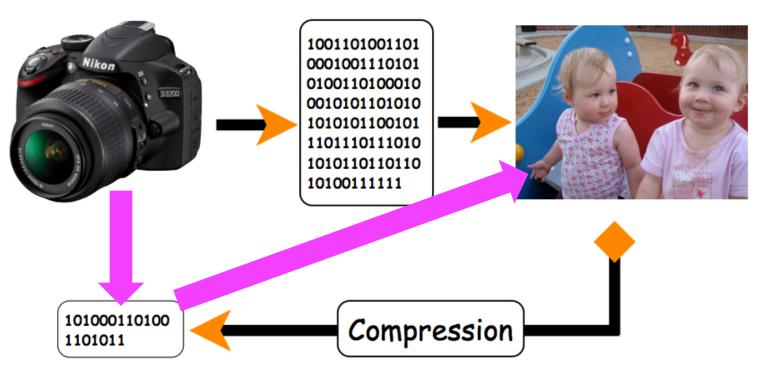




Compressive Sensing/Sampling

□ Standard approach

- First collect, then compress
 - Throw away unnecessary data

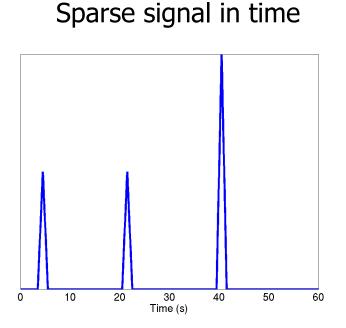


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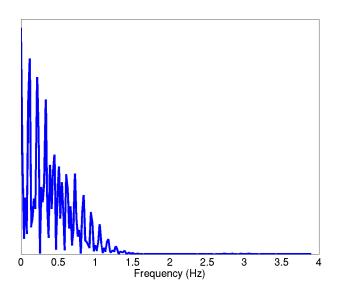


Sample at lower than the Nyquist rate and still accurately recover the signal, and in some cases exactly recover

Sample at lower than the Nyquist rate and still accurately recover the signal, and in some cases exactly recover

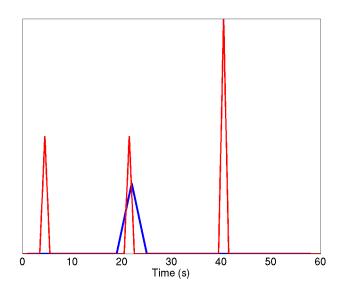






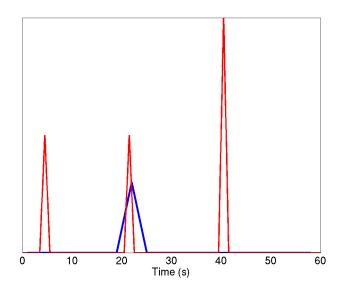
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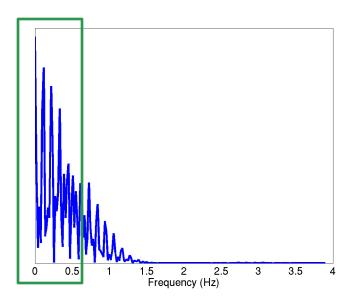
Undersampled in time



Sample at lower than the Nyquist rate and still accurately recover the signal, and in some cases exactly recover

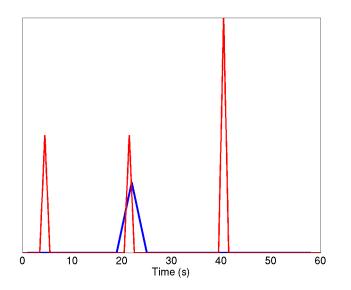
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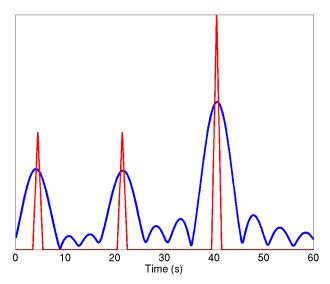


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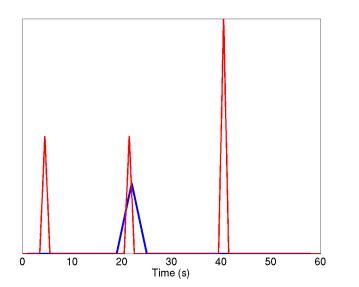


Undersampled in frequency (reconstructed in time with IFFT)

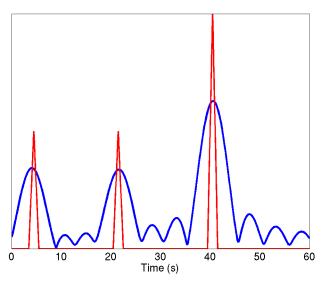


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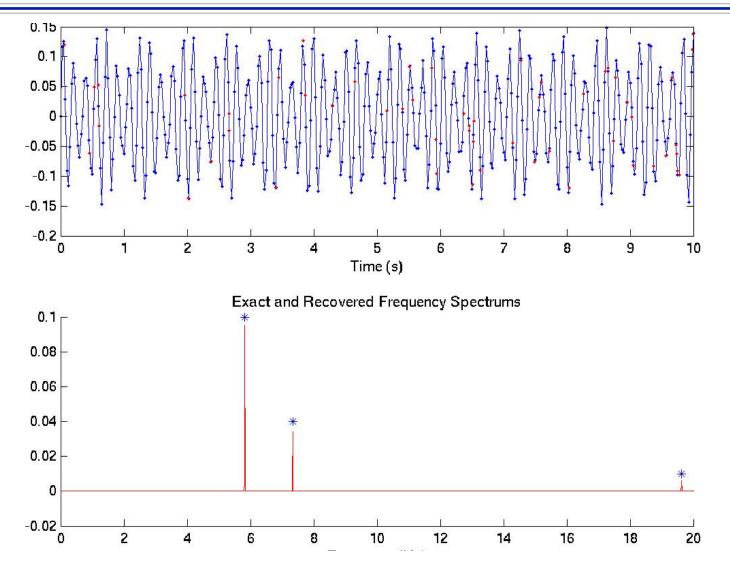


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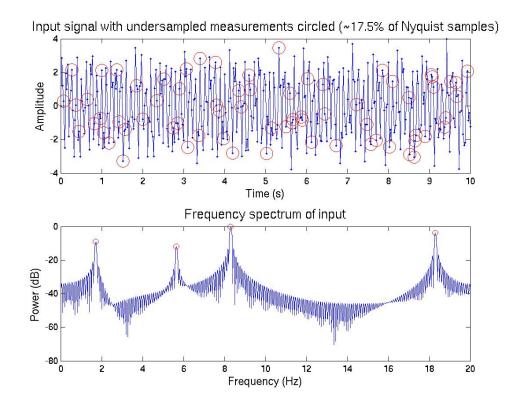


Requires sparsity and incoherent sampling

Compressive Sampling: Simple Example

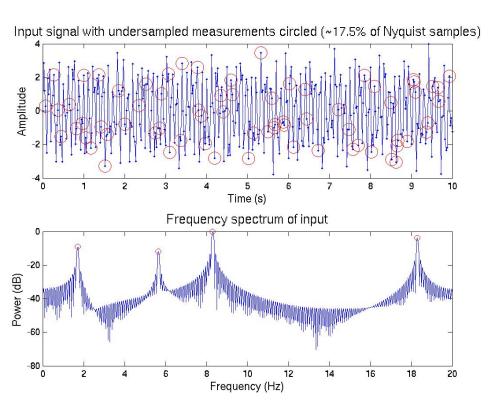






- Sense signal randomly M times
 - $M > C \cdot \mu^2(\Phi, \Psi) \cdot S \cdot \log N$
- Recover with linear program

$$\min \sum_{\omega} |\hat{g}(\omega)| \quad \text{subject to} \quad g(t_m) = f(t_m), \quad m = 1, \dots, M$$

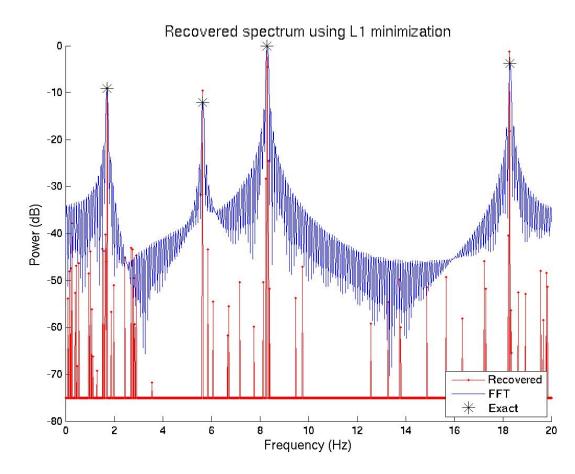


$$\hat{f}(\omega) = \sum_{i=1}^{K} \alpha_i \delta(\omega_i - \omega) \stackrel{\mathcal{F}}{\Leftrightarrow} f(t) = \sum_{i=1}^{K} \alpha_i e^{i\omega_i t}$$

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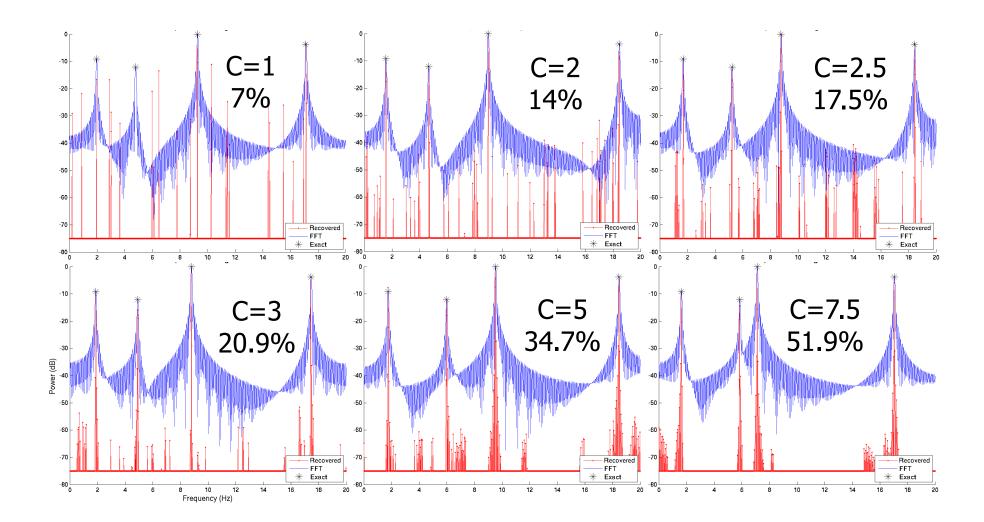
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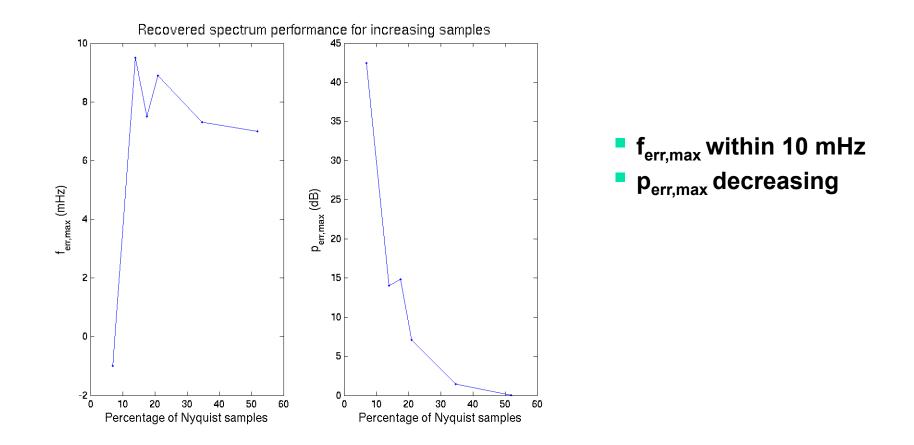


- □ Two relevant "knobs"
 - percentage of Nyquist samples as altered by adjusting optimization factor, C
 - input signal duration, T
 - Data block size

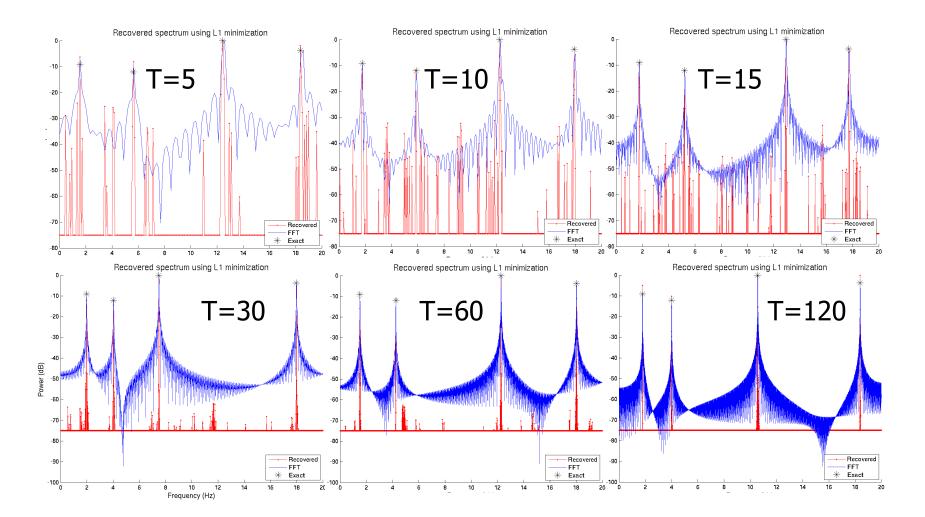




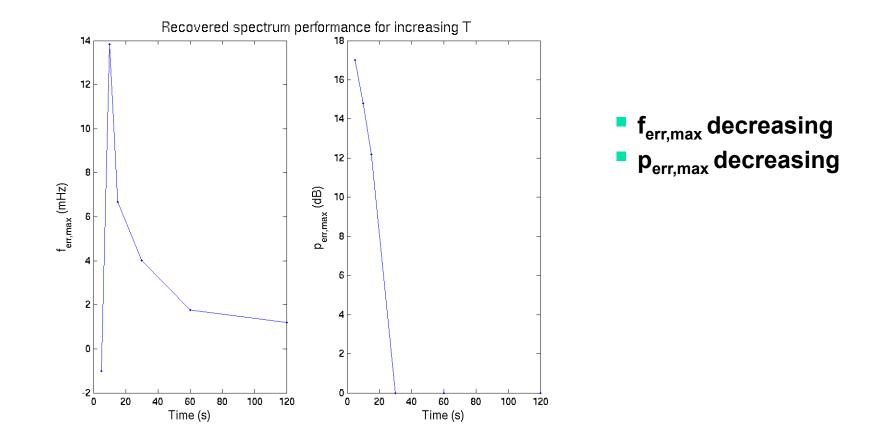




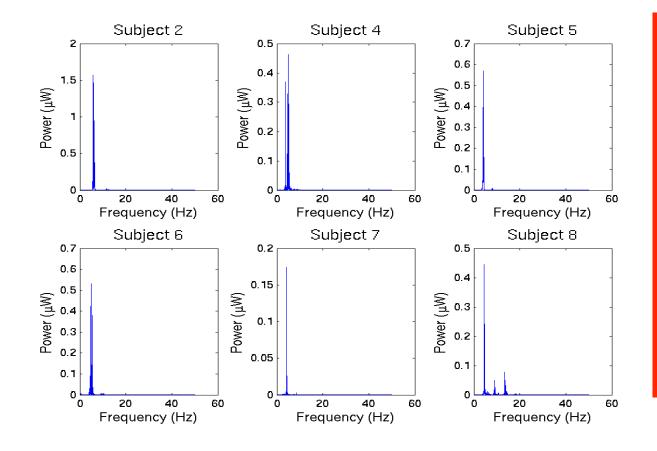








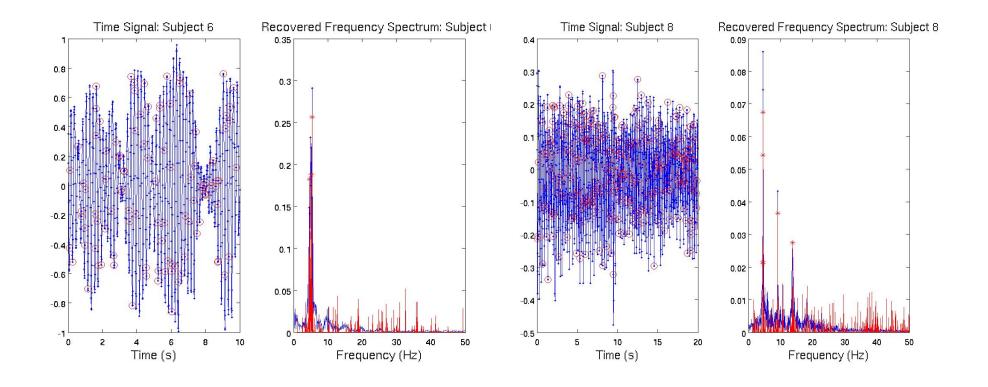
Biometric Example: Parkinson's Tremors



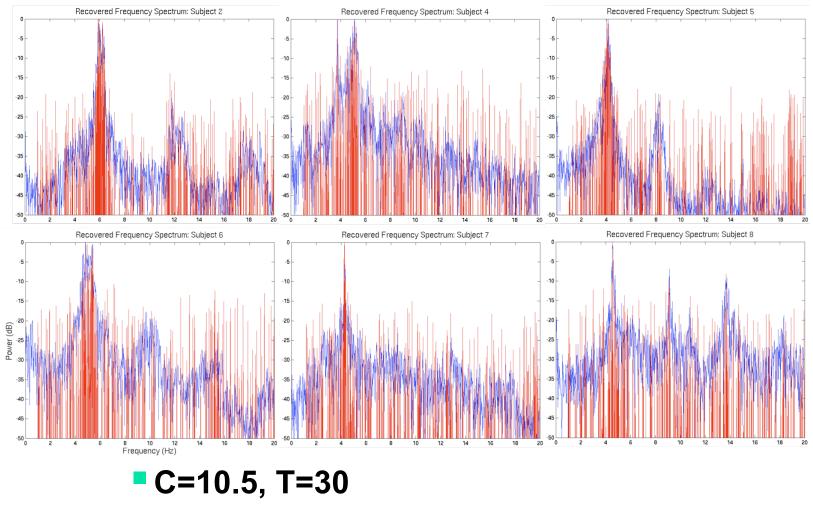
6 Subjects of real tremor data

- collected using low intensity velocity-transducing laser recording aimed at reflective tape attached to the subjects' finger recording the finger velocity
- All show Parkinson's tremor in the 4-6 Hz range.
- Subject 8 shows activity at two higher frequencies
- Subject 4 appears to have two tremors very close to each other in frequency

Compressive Sampling: Real Data

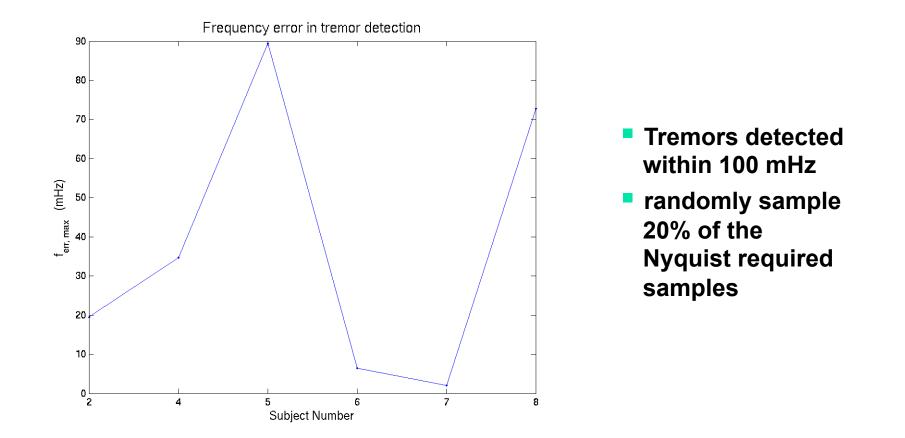






20% Nyquist required samples





Requires post processing to randomly sample!

Implementing Compressive Sampling

- Devised a way to randomly sample 20% of the Nyquist required samples and still detect the tremor frequencies within 100mHz
 - Requires post processing to randomly sample!
- Implement hardware on chip to "choose" samples in real time
 - Only write to memory the "chosen" samples
 - Design random-like sequence generator
 - Only convert the "chosen" samples
 - Design low energy ADC



- Wavelet transform
 - Capture temporal data with fewer coefficients than STFT
- Compressive Sampling
 - Integrated sensing/sampling, compression and processing
 - Based on sparsity and incoherency



- Tania extra office hours
 - Monday 1-3pm
- Tuesday lecture
 - Review
 - Final exam details
- Project
 - Due 4/24